(12) UK Patent Application (19) GB (11) 2 357 409 (13) A

(43) Date of A Publication 20.06.2001

(21) Application No 9929446.4

(22) Date of Filing 13.12.1999

(51) INT CL7
H04H 7/00

(52) UK CL (Edition S)
H4R RSX

(56) Documents Cited
GB 2330751 A GB 2266210 A GB 2255696 A
US 5774566 A US 4939782 A

(58) Field of Search
UK CL (Edition R) H4J JGP JGX, H4R RPX RSX
INT CL7 H03G 5/02, H04H 7/00, H04S 7/00
online: EPODOC,WPI,JAPIO

(71) Applicant(s)
Sony United Kingdom Limited
(Incorporated in the United Kingdom)
The Heights, Brooklands, WEYBRIDGE, Surrey, KT13 0XW, United Kingdom

(72) Inventor(s)
Laurence Kelvin Griffiths

(74) Agent and/or Address for Service
D Young & Co
21 New Fetter Lane, LONDON, EC4A 1DA, United Kingdom

(54) Abstract Title
Audio signal processing

(57) An audio signal processing apparatus (1) is provided with a visual display means (4) coupled to a data processor (2) which operates to generate on the display means (4) a visual representation of a filter characteristic response in the form of an amplitude against frequency curve. A configurable filter (6) is coupled to the data processor (2) which operates to filter audio signals in dependence upon a configurable filter characteristic, and a cursor control device (8) operates in combination with the data processor (2) to control the position and movement of a cursor which is displayed with respect to the amplitude against frequency curve and which serves in use to select and move adjustment points on the curve to adjust the curve in response to user generated commands. The data processor (2) operates to represent visually the adjustments to the amplitude against frequency curve, to adjust correspondingly the filter characteristic response and to generate configuration signals to configure the configurable filter characteristic in accordance with the filter characteristic response. The filter (6) may comprise a plurality of digital filters, e.g. infinite impulse filters. The audio signal processor may also include a data store (114) which provides a facility for other filter characteristic responses to be used to configure the configurable filter. The audio signal processor may operate as a graphic or parametric equaliser.

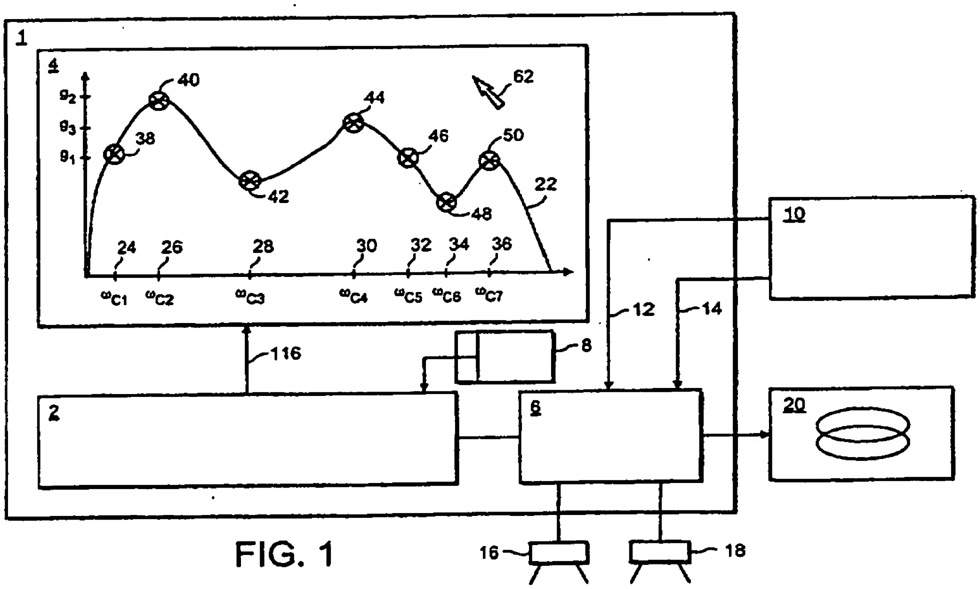
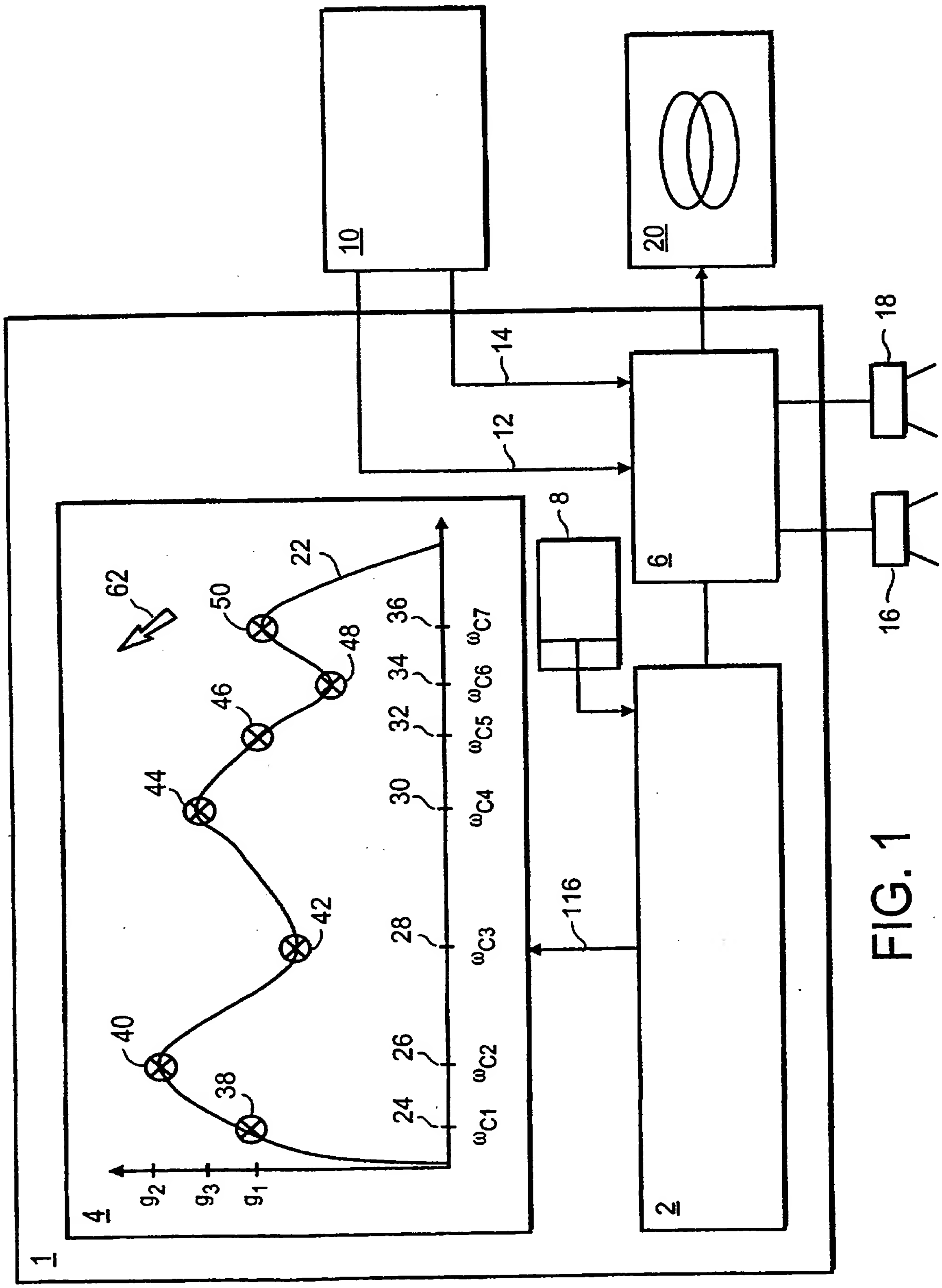


FIG. 1

GB 2 357 409 A

At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.



1
2
4
6
8
10
12
14
16
18
20
22
24
26
28
30
32
34
36
38
40
42
44
46
48
50
62
116
ω_{C1}
ω_{C2}
ω_{C3}
ω_{C4}
ω_{C5}
ω_{C6}
ω_{C7}
g_2
g_3
g_1

FIG. 1

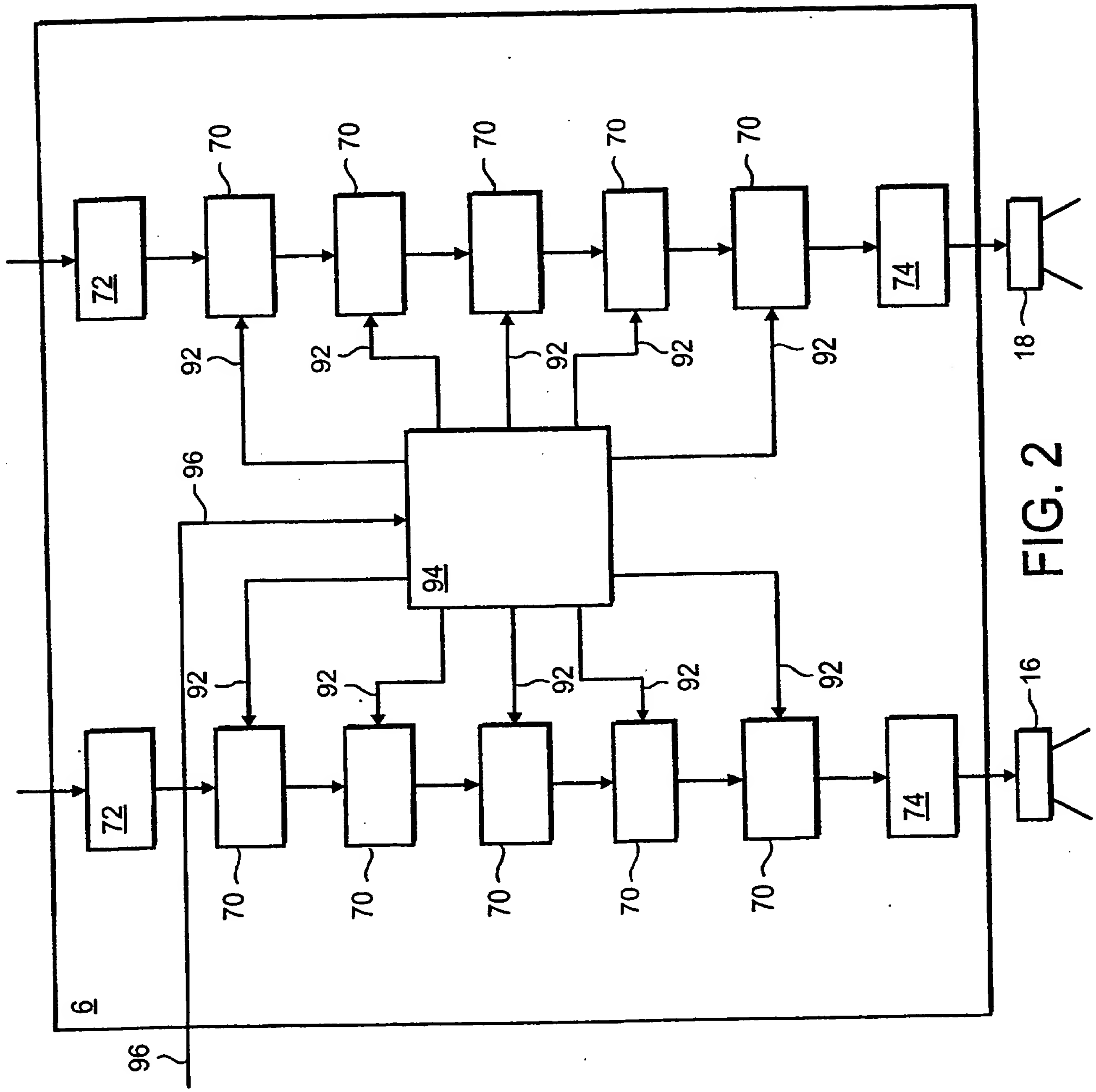


FIG. 2

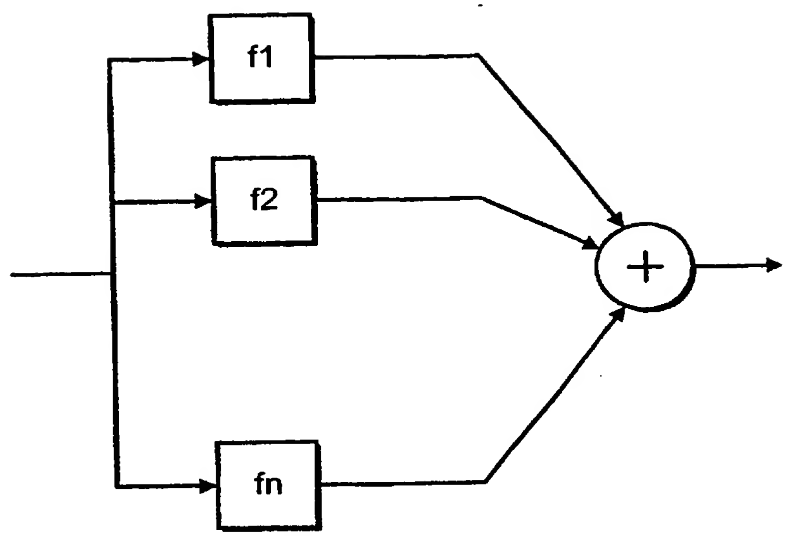


FIG. 3(a)

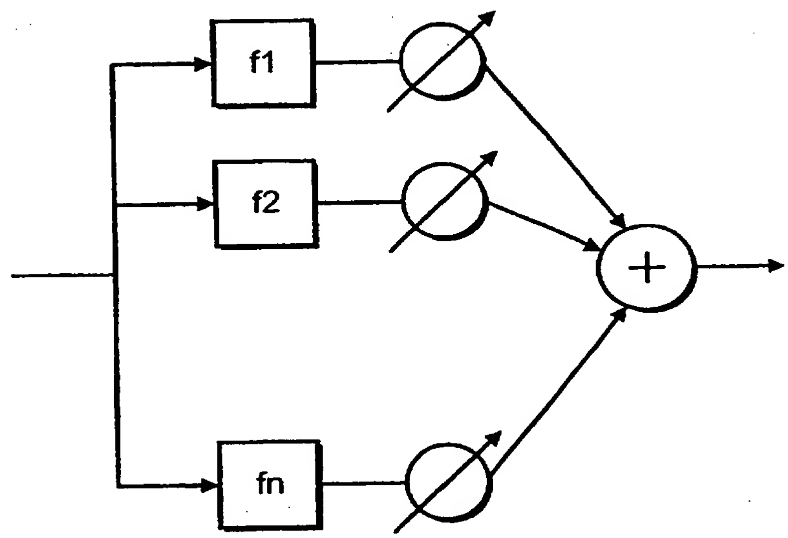


FIG. 3(b)

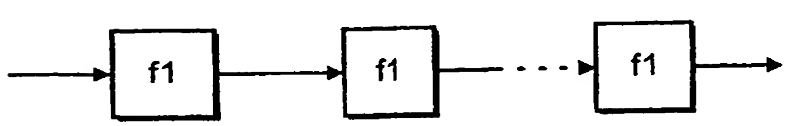


FIG. 3(c)

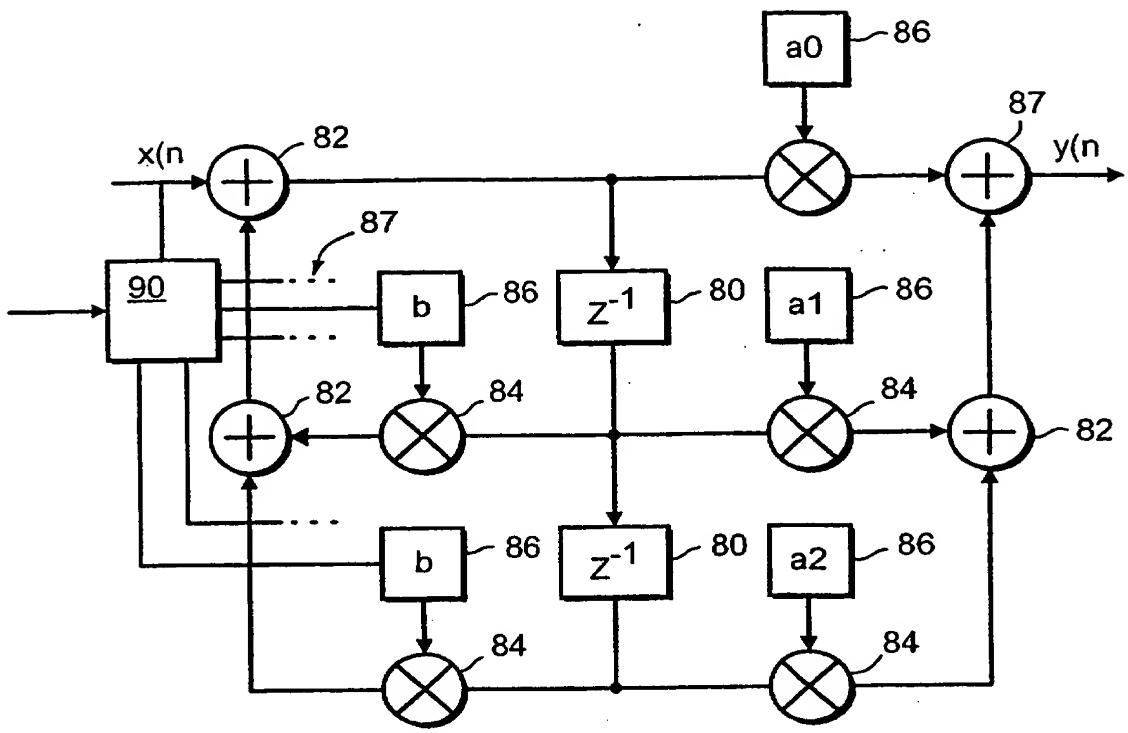

86
a0
87
x(n
82
y(n
87
90
b
86
Z-1
80
a1
86
82
84
84
82
b
86
Z-1
80
a2
86
84
84

FIG. 4

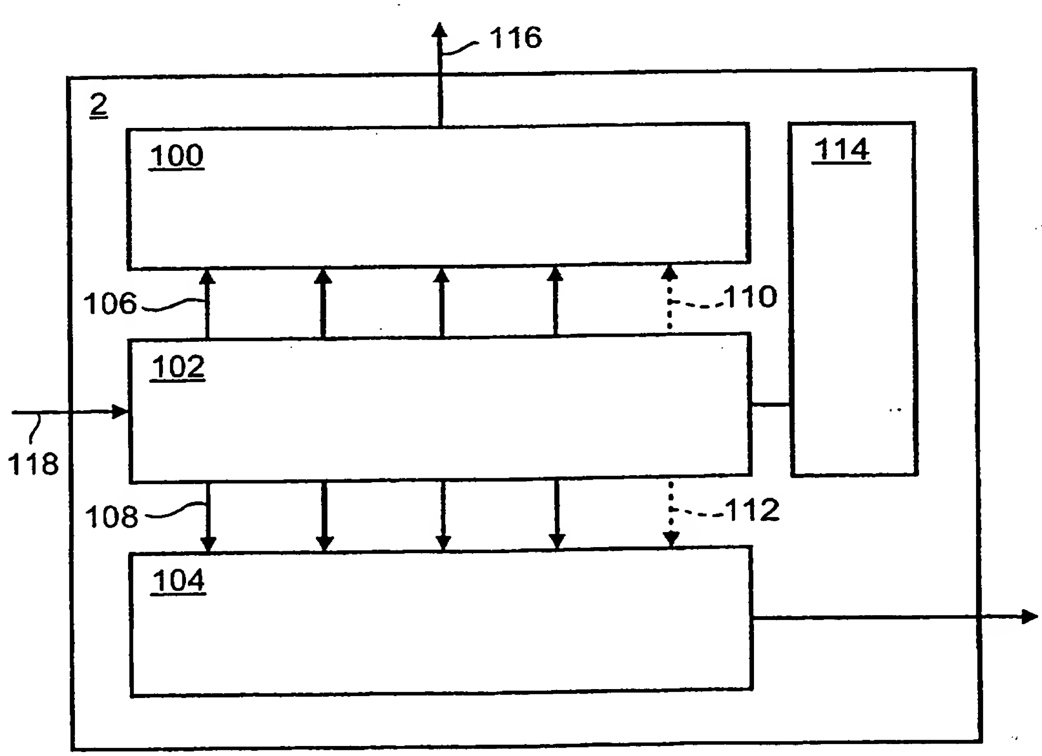

116
2
100
114
106
110
102
118
108
112
104

FIG. 5

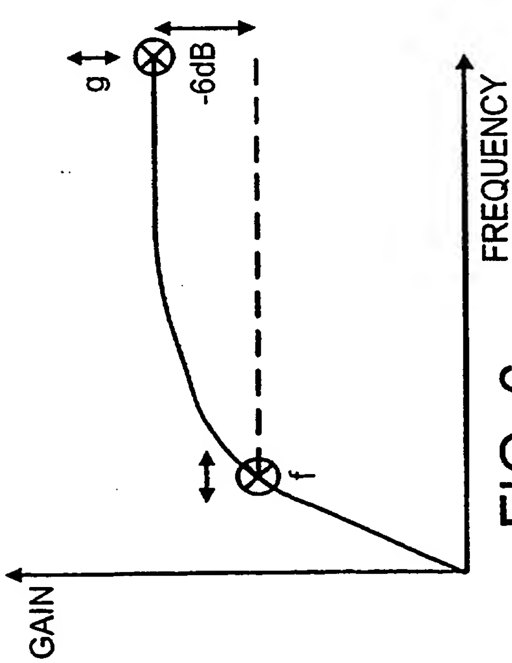


FIG. 6

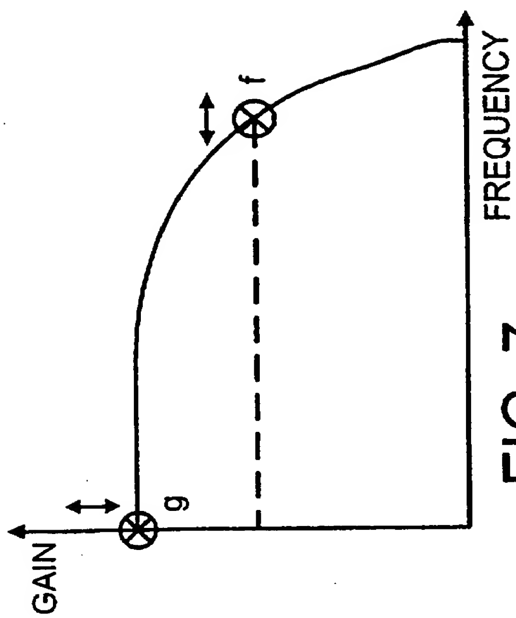


FIG. 7

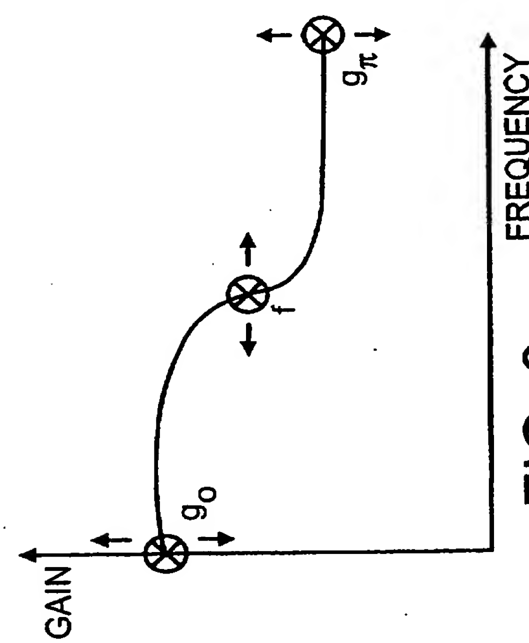


FIG. 8

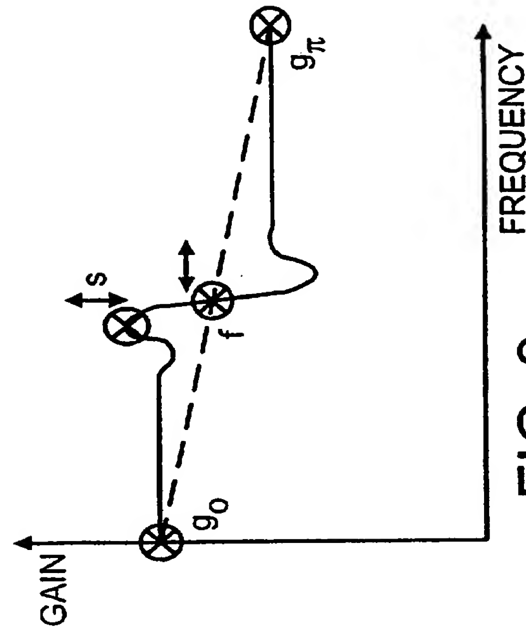


FIG. 9

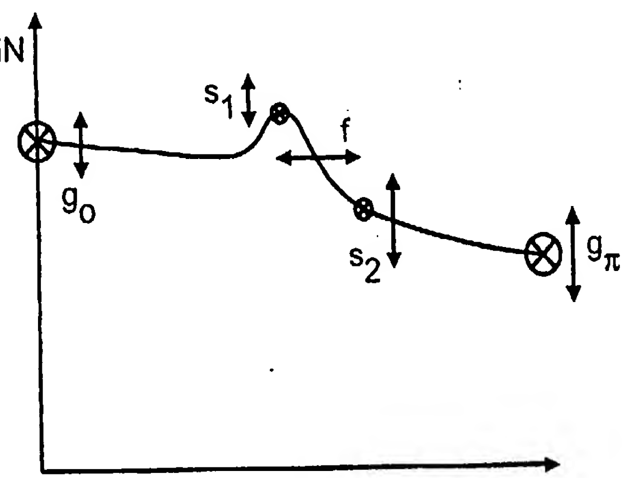


FIG. 10

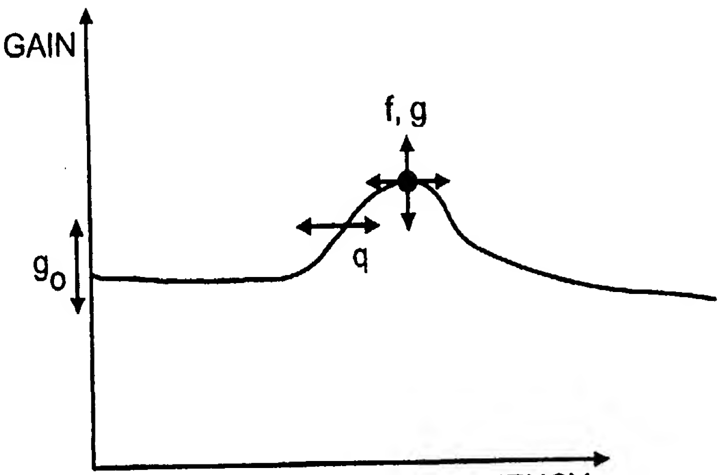


FIG. 11

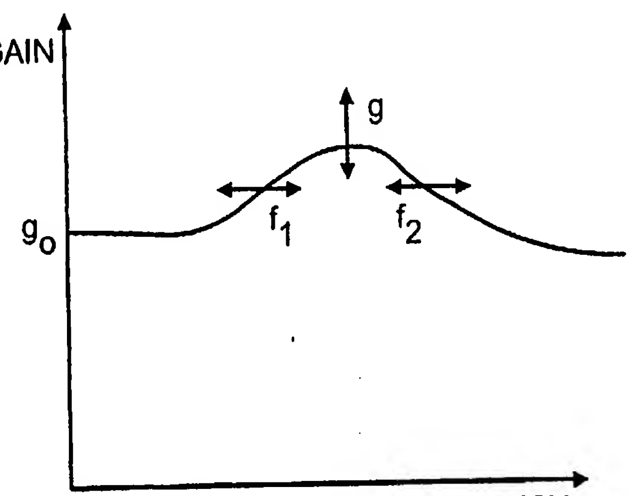


FIG. 12

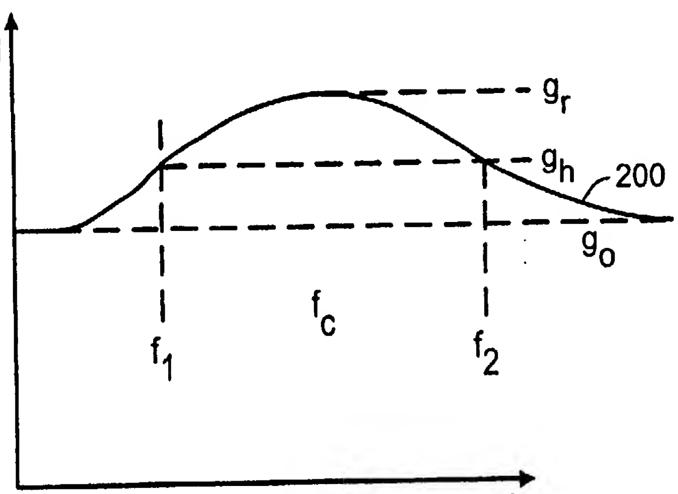


FIG. 13

Signal Processing Apparatus and Method of Processing Signals

The present invention relates to audio signal processing apparatus and methods of processing audio signals.

More particularly, but not exclusively, the present invention relates to audio signal processing apparatus which operate to filter audio signals to the effect of altering the amplitude of the audio signals within each of a plurality of predetermined frequency bands by an amount which may be adjusted individually for each of the frequency bands. Such audio signal processing apparatuses are known to those familiar with the technical field of audio signal processing as graphic or parametric equalisers.

Items of audio reproduction equipment serve to generate an audible reproduction of sound signals which have been recorded, detected or produced in some way. Typically, the reproduced audio signals are amplified and fed to loud speakers which serve to generate audible sound signals for a listener. More particularly, audio reproduction equipment may provide "stereo" reproduction in which a plurality of audio output channels are each fed to a separate loud speaker. In this way, a spatial component is provided to the sound.

Although the audio reproduction equipment is arranged to reproduce the sound signals as accurately as possible, it is often desirable to provide a user of the equipment with a facility for adjusting an amount of attenuation or gain applied to components of the audio signals lying within separate frequency bands. This facility is provided by an audio signal processing apparatus commonly known as an equaliser. The equaliser provides, under user control, a subjective compensation to the audio signals, in dependence upon acoustic and electronic biasing effects introduced by the reproduction equipment and the environment in which the audio signals are reproduced. One example of an equaliser is a "graphic" equaliser. Such graphic equalisers typically include a plurality of adjustable analogue attenuators arranged in a form in which a slider moves within a linear track. Each of the plurality of analogue attenuators is associated with one of a corresponding plurality of mutually exclusive

frequency bands which are arranged to cover the entire band width of the audio signals. The position of the slider within the track determines an amount of attenuation or gain provided to audio signals having frequencies falling within the frequency band with which the attenuator is associated.

Known audio processors which are arranged to perform the function of a graphic or parametric equaliser suffer a disadvantage in that the frequency band width of each of the plurality of frequency bands for which the attenuation of the audio signals may be individually adjusted is fixed. In other words, although the attenuation is adjustable, the frequency range over which this attenuation is applied is not adjustable. A user of the audio equipment is not provided with a facility for adjusting the band width associated with each of the attenuators, or the way in which the audio signals influenced by these attenuators are filtered. Furthermore, the analogue sliders and the filters of the known graphic equalisers suffer degradation over time as a result of dirt and ageing effects to the components which make up the analogue attenuators and corresponding filters. A further disadvantage resides in a requirement for the individually adjusted analogue attenuators to be set for a particular audio signal in accordance with the frequency components of that audio signal. If a different audio signal is reproduced by the reproduction equipment the settings of the analogue attenuators may have to be re-adjusted for the new audio signal, which has a different weighting of signal components with respect to frequency. If the previous audio signal is again reproduced, the settings of the analogue attenuators must be re-established. This represents an inconvenience for the user since it is difficult to reproduce accurately the previous settings of the analogue attenuators, because of the physical constraints imposed by attempting to re-position mechanically each of the analogue sliders or dials within their respective tracks.

According to the present invention there is provided an audio signal processing apparatus comprising, a visual display means, a data processor coupled to the display means which operates to generate on the display means a curve of amplitude against frequency which is representative of a filter characteristic response, a configurable filter coupled to the data processor which operates to filter audio signals in dependence upon a configurable filter characteristic, and a user operated cursor control device

coupled to the data processor which operates in combination with the data processor to control the position and movement of a cursor displayed on the visual display means in response to commands provided by a user, wherein the data processor operates to provide at least one adjustment point on the amplitude against frequency curve which is selectable and moveable in response to the position and movement of the cursor. generate adjustment signals corresponding to selection and movement of the at least one adjustment point by the cursor, represent changes to the amplitude against frequency curve corresponding to the movement of the at least one adjustment point, adjust the filter characteristic response in accordance with the adjustment signals, and generate configuration signals to configure the configurable filter characteristic in accordance with the filter characteristic response.

The filter characteristic response may be in the form of a desired frequency response, that is data representative of a plot of the amplitude of the response with respect to frequency, which may also include a plot of phase with respect to frequency, or the filter characteristic response may be an impulse response of the filter that is to say a time domain response of the filter. Similarly, other ways of representing a characteristic of a filter are included in the term filter characteristic response. By providing an arrangement in which the filter characteristic response is represented on a visual display unit as a curve of amplitude against frequency, and in which a cursor control device can adjust the characteristic filter response by selecting and moving adjustment points on the amplitude against frequency curve and configure a configurable filter in accordance with the filter characteristic response, the audio signal processing apparatus is provided with a means for conveniently and efficiently adjusting the filter characteristic of the configurable filter.

Advantageously, the audio signal processing apparatus may further include a data store which is arranged to store at least one other filter characteristic response. By providing a data store the filter characteristic response which is adjusted and set by a user to a desired response may be stored by the audio signal processing apparatus so that the configurable filter may be reconfigured to that stored filter characteristic response when the audio signals to which the filter characteristic response have been adjusted are once again processed by the audio signal processing apparatus.

Generally, the audio signal processing apparatus is provided with an advantage in having a visual display means and a configurable filter which is configured in accordance with a filter characteristic response which may be adjusted by the user and which is displayed to the user using the visual display means. The audio signal processing apparatus provides an improvement with respect to known audio signal processing apparatuses in that the filter characteristic response may be more easily adjusted and arranged to configure the filters used to filter the audio signals.

Although the configurable filter may be a single filter. for which the filter characteristic response may be a single filter characteristic response, advantageously the configurable filter may be a plurality of configurable filters and the filter characteristic response may provide a characteristic response for each of the configurable filters. Furthermore, each of the filter characteristic responses may be correspondingly represented by one of a plurality of amplitude against frequency curves. In this way, each of the plurality of configurable filters may be assigned to a part of the frequency response of the audio signals and each is individually adjustable by the user.

Each of the plurality of configurable filters may be a digital filter so that the data processor may operate to generate a plurality of filter coefficients, corresponding to the filter characteristic response, for each of the plurality of digital filters. The filter coefficients are therefore represented by and set by the configuration signals.

Although the filter characteristic response for each of the configurable filters may be a high pass filter, a low pass filter, or advantageously a band pass filter having a gain and a centre frequency. In this way each of the configurable filters can be arranged to attenuate signals within an assigned band width which makes up at last part of a range of frequencies present in the audio signals. The band widths assigned to each of the configurable filters may be mutually exclusive with respect to one another and may be arranged together so that the entire frequency response of the audio signals is covered. The data processor may therefore calculate the filter coefficients for the configurable filters from the gain, the centre frequency, and a normalised pass band width determined from the filter characteristic response.

The plurality of configurable filters may be cascaded in series form or arranged in parallel with the output of each parallel filter combined to form a composite output signal.

The term normalised pass band width refers to a difference between a higher and a lower frequency at which the gain in the pass band of the response falls to the same pre-determined level. This level may, for example, be the square root of the gain at the centre frequency of the pass band response.

The plurality of configurable filters may be finite impulse response filters or may be implemented as direct Fourier transform filters in which the audio signals are transformed to the frequency domain and the frequency response of the pass band filters is scaled by the frequency components of the audio signals and the result transformed back to the time domain. However, infinite impulse response digital filters provide an advantage in reducing a number of calculations which are required to implement each of the configurable filters, and also to reduce the number of filter coefficients. To determine the coefficients of the infinite impulse response digital filters, the data processor may operate to determine the gain at zero frequency, Nyquist frequency and a warped frequency determined with reference to the Nyquist frequency.

The data processor may operate to determine which of the filter characteristic responses have changed, to calculate new filter coefficients for each of the digital filters for which the filter characteristic response has changed and to generate configuration signals in accordance with the new filter coefficients. The configuration signals are therefore used to configure those configurable filters which have changed. In this way, only those configurable filters which correspond to the characteristic responses which have changed in accordance with a users desired adjustment are updated with the effect that a number of calculations required to produce the new filter configuration is substantially reduced.

Each of the amplitude against frequency curves may be provided with an adjustment point which facilitates adjustment using the cursor control device. By providing an adjustment point and representing the characteristic filter response as a curve, a user is provided with a convenient means for adjusting not only the gain of the corresponding filter but also the band width since the adjustment points may be

positioned at points, such as half amplitude points, which define the normalised bandwidth of the filter response. Once the curve has been adjusted, the data processor may operate to determine the gain at the centre frequency and the normalised band width from each of the curves from which the filter coefficients are generated.

The data processor may further operate to determine an effective normalised pass band width from an effective filter response, for at least one of the plurality of filters, determine an actual quality factor from the effective normalised pass band width, and re-calculate the filter coefficients and calculate a new gain and/or resonant frequency with the quality factor unchanged.

The quality factor or 'Q' is a known feature of band pass filters which reflects the relative shape of the filter response. A high 'Q' is indicative of a narrow pass band width with respect to frequency, whereas a low 'Q' is indicative of a wide pass band width with respect to frequency. The 'Q' factor is usually, but not exclusively, defined as the ratio of the centre or resonant frequency of the pass band to the difference in frequency between upper and lower points at which the gain at the centre frequency falls to a pre-determined amount. This can be 3dB from the gain at the centre frequency, or can be defined in other ways.

When the coefficients of a digital filter are adjusted, it normally causes an immediate alteration in the value of the output signal, starting with the next sample processed, that is the output signal suffers a sudden jump in value. Such jumps in value cause an unpleasant audio effect known as "zipper noise". To reduce zipper noise to an inaudible level, all coefficients should be altered between two values (from for example 1.0 to 1.3) may be altered by smaller increments (for example) 0.003 for each of a predetermined number of processed samples (for example thousand samples).

Second order digital filters are potentially unstable. If the coefficients are improperly chosen, instead of filtering the sound the filter can generate whistles and other noises. When the coefficients are altered, they must be altered in such a way that the combination of coefficients always produces a stable filter, not only during steady states, but also during adjustments. Transient whistles produced by unstable filters are sometimes known as "birdies".

According to a further aspect of the present invention there is provided a method of processing audio signals comprising the steps of visually representing a curve of amplitude against frequency which is representative of a filter characteristic response, configuring a configurable filter in accordance with the filter characteristic response, filtering audio signals using the configurable filter, providing a visual representation of a cursor which is moveable in response to user generated commands, providing selectable and moveable adjustment points on the amplitude against frequency curve, the adjustment points being selectable and moveable in response to the position and movement of the cursor, adjusting the filter characteristic response in accordance with movement of the adjustment points effected by the cursor, visually representing the adjustment to the filter characteristic response, and re-configuring the configurable filter in accordance with the adjustment to the filter characteristic response.

Embodiments of the invention will now be described, by way of example only, with reference to the accompanying drawings in which:

Figure 1 is a schematic block diagram of an audio signal processing apparatus according to the present invention;

Figure 2 is a schematic block diagram of a configurable filter forming part of the audio signal processing apparatus shown in Figure 1;

Figure 3(a) is a schematic block diagram of an arrangement of filters in parallel, 3(b) is the arrangement of Figure 3(a) with attenuators adjusting the filter outputs and Figure 3(c) is a schematic block diagram of a concatenated series of filters;

Figure 4 is a schematic block diagram of one of a plurality of configurable filters which make up the filter shown in Figure 2;

Figure 5 is a schematic block diagram of a data processor which forms part of the audio signal processing apparatus shown in Figure 1;

Figure 6 is a graphical representation of a plot of gain against frequency for a high pass filter;

Figure ~~6~~ 7 is a graphical representation of a plot of gain against frequency for a low pass filter;

Figure 8 is a graphical representation of a plot of gain against frequency for a shelf filter;

Figure 9 is a graphical representation of a plot of gain against frequency for a symmetrical shelf filter;

Figure 10 is a graphical representation of a plot of gain against frequency for an asymmetrical shelf filter;

Figure 11 is a graphical representation of a plot of gain against frequency with handles for adjusting a resonance filter;

Figure 12 is a graphical representation of a plot of gain against frequency for the plot of Figure 11, with other adjustment handles; and

Figure 13 is a graphical representation of the plot of the resonance filter shown in Figures 11 and 12, illustrating the calculation of coefficients for the configurable filter.

As already explained, the term graphic or parametric equaliser is used to describe a signal processing apparatus used with audio reproduction equipment to filter different parts of a frequency spectrum of an audio signal by an adjustable amount in order to achieve a desired audio effect. An audio signal processing apparatus representing an example embodiment of the present invention is shown in Figure 1. In Figure 1 an audio signal processing apparatus 1 is shown to comprise a data processor 1 connected to a visual display unit 4 and to a configurable filter 6. Also coupled to the data processor 2 is a user operated cursor control device 8, which may for example be a computer mouse or the like. Also shown in Figure 1 is an audio signal generator which may be for example an audio reproduction apparatus such as a hi fidelity stereo or apparatus for producing an audio sound track. The audio signal generator 10 generates audio signals on two output channels 12, 14 which are representative of left and right stereophonic signals. The left and right channels from the audio signal generator 10 are fed to respective loud speakers 16, 18 via the configurable filter 6 of the audio signal processing apparatus 1. The loud speakers provide a convenient way to evaluate the effect of the filter, but are not otherwise necessary. Furthermore, Figure 1 includes a recording device 20 which is coupled to a further output from the configurable filter 6.

In operation the audio signals generated by the audio signal generator 10 are filtered by the configurable filter before being converted into an audible signal by the loud speakers 16, 18. The audio signal processing apparatus 1 may also include other components such as amplifiers, which are not shown in Figure 1. The configurable filter 6 is configured in accordance with a filter characteristic response fed to the configurable filter by the data processor 2. The data processor 2 furthermore operates to generate a visual representation of the filter characteristic response as a curve or curves of amplitude against frequency displayed on the visual display unit 4. An example of a representative curve 22 is shown in Figure 1 within the visual display unit 4. The example the filter characteristic response 22 is represented as a relationship of amplitude against frequency, although it will be appreciated that other responses are possible. Furthermore, there may be a plurality of curves corresponding to band pass filters, having band widths which are mutually exclusive with respect to the other band widths, which make up the frequency range of the audio signals generated by the audio signal generator 10. As will be appreciated however, the bandwidths of each of the responses need not be mutually exclusive and may overlap or coincide as shown in the example response. As shown in Figure 1 within the display unit 4 the curve is provided with a frequency points 24, 26, 28, 30, 32, 34, 36 ω_n and gain at the frequency point g_n. The response may also include the frequency of each of other points on the curve corresponding to a gain which is half that at the centre frequency ω_{cn}, where n is an index designating the filter producing the response number. Furthermore, the frequency gain points on the curve 22, are provided with a visual representation of adjustment points which are displayed in the form of handles. The handles provided to provide reference points which can be dragged so that the filter characteristic responses can be changed to achieve a desired audio effect. This is facilitated by use of the computer mouse 8 which operates to convert movement in an x-y plane into movement of a cursor 62, which is also displayed by the data processor on the visual display unit 4. By appropriately selecting buttons on the computer mouse in a conventional way, the handles may be selected and dragged to effect a desired adjustment to the amplitude against frequency curve which is correspondingly reflected in an adjustment to the filter characteristic response. That is, the signals generated by

the computer mouse are interpreted by the data processor 2, to the effect of making a corresponding adjustment to the amplitude against frequency curve which is then displayed to the user on the visual display unit 4. Correspondingly, the filter characteristic response associated with the curve is adjusted to reflect the change to the amplitude against frequency curve. The data processor then operates to generate signals which are fed to the configurable filter 6 which are representative of the adjusted filter characteristic response and which serve to configure the filter 6 in accordance with the amplitude against frequency curve displayed on the visual display unit 4. Thus, when one of the characteristic responses represented by the curve 22 is adjusted, the data processor operates to generate signals which are representative of this adjustment and which are used to configure the configurable filter 6. The configurable filter 6 then operates to filter the audio signals from the audio signal generator 10 in accordance with the filter characteristic response.

The configurable filter 6 is shown in more detail in Figure 2 where parts also appearing in Figure 1 bear identical numerical designations. In Figure 2 the filter 6 is shown to be made up from a plurality of configurable filters 70 which are divided respectively into two parts, the first part being associated with the left channel 12 from the audio signal generator and the second half being associated with the right channel 14 from the audio signal generator. Effectively the first and second halves of the configurable filters are performing the same operations to the audio signals provided on the left and right channels of the audio signal generator. Since these same operations are repeated in both the left and right halves, only the operations with respect to the first set of configurable filters will be described. The first set of configurable filters is shown to comprise five configurable filters each of which corresponds to one of the filter characteristic responses which serve to make up the response displayed as the corresponding amplitude against frequency curve 22 on the visual display unit 4. Thus the first configurable filter is arranged to perform filtering of the audio signals from the audio output channel 12 in accordance with the first curve of the filter characteristic response. Correspondingly, the second configurable filter performs the filtering operation in accordance with the second curve 24, the third configurable filter performs a filtering operation in accordance with the third curve 26,

and correspondingly the fourth and fifth configurable filters operate to filter the audio signals in accordance with the fourth and fifth frequency curves 28, 30.

The configurable filters shown in Figure 2 form a concentrated series. However, the plurality of configurable filters may be concatenated in series or combined in parallel. A parallel arrangement of filters is shown in figure 3(a). In figure 3 (a) each of the filters f_1 f_n must have precisely the same phase characteristics, otherwise the final mixing can cause spurious augmentation or cancellation at particular frequencies. In practice this can be achieved digitally using FIR (Finite Impulse Response) filters. The characteristics of such a filter can be smoothly adjusted by putting an attenuator in front or behind to simply control how much of that signal is mixed into the output. This arrangement is shown in figure 3(b). The bandwidth characteristics of f_1 to f_n are calculated once at design time and there after fixed.

Infinite impulse response (IIR) filters can be adjusted 'on-the-fly'. These have to be connected in series to avoid phase cancellation. This is illustrated in Figure 3(c), which illustrates a series concatenation of IIR filters.

In the parallel case the resultant characteristic becomes $f_1a_1 + f_2a_2 +$ $+ f_na_n$.
In the concatenated series case the resultant characteristic becomes f_1 x f_2 x f_3
. The arrangement shown in Figure 2 must accommodate in the series case a situation in which, if f_1 knocks out same frequency, the other filters cannot put it back.

In the example embodiment of the present invention shown in Figure 2, each of the configurable filters 70 is an IIR digital filter. However, as explained, the configurable filters could be finite impulse response filters or indeed the filter characteristic response could be represented in the frequency domain, and the convolution of the audio signals with the impulse response of the filter could be performed in the frequency domain by converting the audio signals into the frequency domain, using a discrete Fourier transform, multiplying with the frequency domain filter response and converting the result of the multiplication back into the time domain. Infinite impulse response filters provide a particular advantage in that the rate of attenuation or gain with respect to frequency in relation to the number of calculations performed to effect the filtering operation is substantially greater than that

which can be achieved with other filter types, and provide a smooth response which can be flexibly readjusted.

In order to filter the audio signals produced at the output channels 12, 14, the analogue audio signals are converted into digital form using analogue to digital converters 72. Correspondingly the output from the final configurable filter is converted into the analogue domain using a digital to analogue converter 74. Alternatively however, the audio signal processing provided by the configurable filter could be applied to the audio signals in the digital domain, obviating a requirement for the analogue-to-digital and digital-to-analogue converters 72, 74.

An example of an infinite impulse response digital filter is shown in Figure 4 where parts also appearing in Figure 2 bear identical numerical designations. The filter shown in Figure 4 is a second order filter and is represented in a form known as the Direct Form II having a minimum number of delay elements as described in a publication entitled "Digital Signal Processing" by A Oppenheim and R Schafer published by Prentice Hall International Inc. 1975 ISBN No. 013-214107-8. As shown in Figure 4, the infinite impulse response filter has delay elements 80, adders 82, and multipliers 84. The input signal is represented as x(n) whereas the output signal samples are represented as y(n). Operatively coupled to each of the multipliers 84 is a data register 86 which serves to store and feed to the multiplier a coefficient which is used to multiply the signal received at an input of the multiplier at a corresponding point in the filter network. As shown in Figure 4 each of the data registers 86 includes a coefficient a_0, a_1, a_2, b_1, b_2, the coefficients being arranged to implement the response of the infinite impulse response filter. These coefficients are loaded via conductors 88 from a filter controller 90. The filter controller 90 updates the registers 86 between clock cycles determined in accordance with the rate at which input samples x(n) are received at the filter. Thus in between samples, the filter controller operates to update the coefficients so that for a subsequent sample the new filter coefficients are used. Advantageously, such updates are done with a small adjustments to each coefficient each cycle to avoid "zipper" noise. Furthermore such updates are performed so that at every stage the combination of the five coefficients results in a stable filter. This avoids "birdies" noise.

The coefficient processor 94 shown in Figure 2 operates to feed coefficients received at an input 96, which is connected to the data processor 2 of Figure 2, to respective ones of the configurable filters 70. Therefore the configuration signals are fed from the data processor 2 via a conductor 96 to the coefficient processor 94. The configuration signals are representative of the filter coefficients and are generated in accordance with each of the respective filter characteristic responses represented as the amplitude against frequency curves shown in the display unit 4. The coefficient processor 94 therefore serves to identify which of the coefficients should be fed to respective ones of the configurable filters 70 via the conductors 92. In this way the filter 6 is configured in accordance with the filter characteristic response adjusted as required by the user.

The operation of the data processor 2 is shown in more detail in Figure 5 where parts also appearing in Figure 1, 2 and 4 bear identical reference numbers. In Figure 5 the data processor 2 is shown to comprise a video driver 100 which is coupled to a filter characteristic processor 102. Coupled to the filter characteristic processor 102 is a filter coefficient generator 104. As an illustrative example to correspond with the amplitude against frequency curve 22 displayed on the visual display unit 4, the video driver 100 and the filter coefficient processor 104 are coupled to the filter characteristic processor 102 by five conductors 106, 108 to produce five configurable filter responses which together form the filter characteristic curve 22. However as will be appreciated, further amplitude against frequency curves could be provided and correspondingly further configurable filters could be present in the filter 6. To reflect this, further dashed lines 110 and 112, illustrate that further configurable filters and responses could be present.

As shown in Figure 5, a data store 114 is also coupled to the filter characteristic processor 102. The data store serves to store different filter parameters with which the audio filter 6 can be configured. However, the filter characteristic processor 102 operates at any one time with one active filter characteristic response. The active filter characteristic response, which comprises parameters for each of the filters, is represented visually on the visual display unit 4 by feeding each of these responses via the conductors 106 to the video driver 100. The video driver 100 operates to generate

signals which produce amplitude against frequency curves representative of the filter characteristic responses, and which are arranged in a form in which they can be displayed on the visual display unit 4. These signals are fed by the video driver to the visual display unit 4 via the output conductor 116. Also fed on a further input to the filter characteristic processor 102 are the adjustment signals generated by the computer mouse 8. The command signals fed from the mouse 8 are interpreted by the filter characteristic processor 102 in combination with the video driver 100 in order to both represent the adjustment to the amplitude against frequency curve 22 shown on the display unit 4, and to correspondingly adjust the filter characteristic response. Once an adjustment has been effected, the filter characteristic processor 102 operates to identify which of the filter characteristic responses, which make up the curve 22 have changed. Those responses which have changed are then fed to the filter coefficient processor 104 via the conductors 108. The filter coefficient processor 104 then operates to generate a new set of filter coefficients representative of the adjusted filter characteristic response, corresponding to the adjusted curve. The new filter coefficients are thereafter fed via conductor 96 to the coefficient processor 94 within the audio filter 6. By identifying those characteristic responses which have changed from those which have not changed, a saving in terms of a number of calculations required to generate the filter response is made in that only those characteristic responses which have changed are correspondingly updated. A further advantage is provided to the audio signal processor 1 in that the computer mouse 8 can be arranged to indicate which of the filter characteristic response stored in the data store 114 is to be made the active filter characteristic response. In this way, the filter coefficient processor operates to generate filter coefficients for each of the configurable filters 70 in accordance with the active filter characteristic responses which are once again fed via the conductor 96 to the coefficient processor 94 at which point they are distributed to respective configurable filter 70. Once the configurable filters within the filter 6 have been configured, the audio signals from the audio signal generator 10 after filtering can be recorded as desired, using the recording device 20, under control of the data processor 2.

The operation of the filter coefficient processor 104 will now be described in more detail. In the example embodiment described above, each of the configurable

filters are second order IIR filters, so that the filter coefficient processor operates to generate the coefficients a_0, a_1, a_2, b_1, b_2 of the filter shown in Figure 4. However it will be appreciated that the operation of the filter coefficient processor can be extended to other filter orders, requiring a greater or lesser number of filter coefficients.

There are seven different kinds of filters. These are illustrated in Figures 6, 7, 8, 9, 10, 11 and 12. One example of these filters, known as the resonant filter poses a special problem. If the centre frequency and/or gain is varied then it is desirable to keep the filter about as sharp as before. In this case, an approximation for the Q factor of the filter is calculated from the existing parameters and is then preserved as the frequency against gain characteristic is altered. The approximation used is well behaved. For instance, if one attempts to preserve the exact Q i.e. to preserve $\frac{f}{f_2 - f_1}$ and one increases f, then, for low Q there comes a point where f_1 must exceed the Nyquist frequency and/or f_2 be less than zero. The alternative "approximate" expression under these circumstances is a very bad approximation, but is very well behaved. The resonant filter coefficient calculation process is the only circumstance when this approximation for Q is used.

In general we calculate different variables for the different kinds of filter. The useful quantities are;

g_0 the gain at "D.C." (zero frequency),

g_π the gain at the Nyquist frequency,

g_{res} the gain at resonance (resonant filter only),

t warped frequency - at resonance for a resonant filter,
- at mid point for a shelf filter,
- at 3dB cut for a low and high filter,
- at centre point and at high and low frequency half-effect points for resonant filters.

The parameters are therefore;

Resonant:	g_π	g_{res}	f	f_1 or f_2
Resonant[1]:	g_π	g_{res}	f_1	f_2
Low (high pan):	g_π		f	

High (low pan)	g_o		f		
Shelf	g_0	$g\pi$	f	s	
Symmetric shelf	g_0	$g\pi$	f	s	
Asymmetric shelf	g_0	$g\pi$	f	s_1	s_2

The parameters s (s1, s2) are controlled by dragging adjustment handles away from the gain against frequency plot. For a resonant filter

if

$g_{res} > g\pi$ then

$g_p = w/(2\ q\ \sin\omega)$

$g_z = g_p\ G_{lin}$

else

$g_z = w/(2\ q\ \sin\omega)$

$g_z = g_z/G_{lin}$

For simple shelf $\quad g_z = g_p = 1/\sqrt{2}$

For symmetric shelf $\quad g_z = g_p = s/\sqrt{2}$

For asymmetric shelf $\quad g_z = s_1/\sqrt{2},\ g_p = \dfrac{s_2}{\sqrt{2}}$

Further high or low pass $\quad g_z = 0,\ g_p = \dfrac{1}{\sqrt{2}}$

For notch $\quad g_z = 0,\ g_p = \dfrac{\left|t^2 - d^2\right|}{2td}$

where d = warped half gain frequency

For resonant $\quad t_z = t_p = \tan\dfrac{w}{2}$

For resonant[1] $\quad t_z = t_p = \sqrt{v_1 v_2}$

where $\quad v_1 = \tan\dfrac{w_1}{2},\ v_2 = \tan\dfrac{w_2}{2}$

$\quad w1 = \dfrac{2\pi f1}{F_n},\ w_2 = \dfrac{2\pi f2}{F_n}$

For shelf (all varieties) $t_z = \left(\frac{g_o}{g_\pi}\right)^{1/4} \tan\frac{w}{2}$

$t_p = \left(\frac{g_\pi}{g_o}\right)^{1/4} \tan\frac{w}{2}$

For low pass t_z = "∞" (e.g. 10^6)

$t_p = \tan\frac{w}{2}$

For higher pass $t_z = 0$

$t_p = \tan\frac{w}{2}$

There is an overall correction factor for the calculated characteristic which is calculated as follows:

Resonant shelf, low pass, notch: $A = g_\pi \dfrac{\left(1 + 2g_z t_z + t_z{}^2\right)}{\left(1 + 2g_p t_p + t_p{}^2\right)}$

High pass: $A = g_o \dfrac{t_p{}^2}{\left(1 + 2g_p t_p + t_p{}^2\right)}$

An example of filter characteristic response which will be used to explain the operation of the filter coefficient processor is shown in figure 13. In figure 13, a curve 200,. represents a response of gain against frequency. In Figure 13 a centre frequency f_C and the frequency of the respective half gain values f_1 and f_2 are shown. The response to parameterised in terms of f, f_l and f_h, g_r and g_o, in which f_l, f_r and f_h are interdependent terms, and f_l and f_h are the lower an upper "half effect" points chosen so that $g_n = \sqrt{g_o g_r}$. This means that if g_o= I (i therefore OdB) and g_r = 2dB then g_h = 1dB. The traditional "half the gain" approach breaks down in this case because there is no frequency at which the gain is half the gain at resonance, i.e. where g_r is 3dB less than g_r. Furthermore the gain at the centre frequency g_C is shown. This centre frequency f_C is then calculated as an angular frequency with respect to the sampling frequency F_S according to equation (1) below. In order to calculate the co-ordinates of the poles and zeros of the filter, this angular frequency is then represented using a

warped frequency t calculated from equation (2) below. Another variable used to determine the co-ordinates of the poles and zeros is G_{lin}, which is calculated from the gain at the centre frequency of the pass band g_{cn}, from equation (3). A remaining influence on the filter coefficients is the quality or 'q' factor of the resonant frequency which is representative of the pass band response. This is calculated using equation (4) from the half gain frequency values f_1 and f_2. In equations (1) to (6) below, n is the number of the filter response.

A co-ordinate representing the gain of the zero of the second order filter is calculated from equation (5) which provides g_{zn}. Similarly a co-ordinate of the gain of the pole of the filter g_{pn} is calculated from equation (6).

$$\varpi_{cn} = 2\pi f_{cn} / F_S \qquad (1)$$

$$t_n = \tan(\varpi_{cn} / 2) \qquad (2)$$

$$G_{lin} = 10 g_c / 20 \qquad (3)$$

$$q_n = \frac{\varpi_c}{2g_c \sin(\varpi_c)} \cong \frac{f_c}{f_1 - f_2} \qquad (4)$$

$$g_z = \omega / (2q \sin(\varpi_c)) \qquad (5)$$

$$g_p = \omega / (2q \sin(\varpi_c)) / G_{lin} \qquad (6)$$

Having calculated the co-ordinates of the pole and zero of the second order filter, the coefficients of the infinite impulse response filter to be configured are determined from equations (7) to (11) below, which use the terms of the co-ordinates g_z, g_p and t.

$$a_0 = A \qquad (7)$$

$$a_1 = 2A\,(t_z^2-1) / (1 + 2\,t_z g_z + t_z^2) \qquad (8)$$

$$a_2 = A\,(1-2t_z g_z+t_z^2) / (1 + 2\,t_z g_z+t_z^2) \qquad (9)$$

$$b_1 = -2(t_p^2-1) / (1 + 2\,t_z g_z+t_z^2) \qquad (10)$$

$$b_2 = -(1-2\,t_z g_z+t_z^2) / (1 + 2\,t_z g_z+t_z^2) \qquad (11)$$

If the peak gain or peak frequency of a resonant filter is adjusted, then it is necessary to reverse the normal order of calculation to find the half-effect points (this

time exactly) so as to be able to mark them on the graph. In this way the "approximate Q" may be preserved. Equations (12) and (13) give then warped half-effect frequencies. Then

$$f_1 = \frac{F_N}{\pi} \arctan(v_1) \tag{12}$$

$$f_2 = \frac{F_N}{\pi} \arctan(v_2) \tag{13}$$

If the half-effect frequencies are given then the exact Q is

$$\frac{(f_2 - f_1)}{f_{res}} = \text{Qact} \tag{14}$$

and then the warped frequency at the half effect points becomes:

$$v_1 = \tan\left(\frac{\pi f_1}{F_n}\right) \text{ warped lower half effect point}$$

$$v_2 = \tan\left(\frac{\pi f_2}{F_n}\right) \text{ warped upper half effect point}$$

$$t = \sqrt{v_1 v_2}$$

g_z and g_p can then be calculated from equations 15 and 16

$$g_z = \frac{(l + t_2)}{2t} \tan\left(\frac{w_c}{2Qact}\right)\sqrt{G_{lin}} \tag{15}$$

$$g_p = \frac{(l + t_2)}{2t} \tan\left(\frac{w_c}{2Qact}\right)\sqrt{G_{lin}} \tag{16}$$

For a resonant filter the gain at zero frequency, g_o = the gain at the Nyquist frequency g_π. The half-effect point is where the gain is $\sqrt{(g_{res} / g)}$.

Any form of filter response may be used to represent the desired filter characteristic, and although the embodiment has been described with a plurality of characteristic responses to be used in configuring a corresponding plurality of configurable filters, it will be appreciated that the filter characteristic could be implemented directly with a single configurable filter. In this case at least one adjustment point is provided on the curve representing the amplitude against frequency response, to adjust the filter characteristic response using the computer mouse.

There are other ways of effecting a filter operation. Furthermore filters other than second order may be used. As will also be appreciated, although the data processor has been described as having separate processors to perform the functions of the video driver, the filter characteristic processor and the filter coefficient processor, it will be understood that these could be implemented entirely in software or as a combination of hardware and software with each of the video driver 100 characteristic processor 102 and filter coefficient processor 104, being implemented as a separate processing board with associated processing software.

Since the audio signal processing apparatus can be implemented as a software computer program running on a computer, it will be understood that a computer program embodying the steps of the method of processing audio signals according to the invention, and a computer running such a program fall respectively within the scope of the invention.

CLAIMS

1. An audio signal processing apparatus comprising,
- a visual display means,
- a data processor coupled to said display means which operates to generate on said display means a curve of amplitude against frequency which is representative of a filter characteristic response,
- a configurable filter coupled to said data processor which operates to filter audio signals in dependence upon a configurable filter characteristic, and
- a user operated cursor control device coupled to the data processor which operates in combination with said data processor to control the position and movement of a cursor displayed on said visual display means in response to commands provided by a user, wherein the data processor operates to
- provide at least one adjustment point on said amplitude against frequency curve which is selectable and moveable in response to the position and movement of said cursor,
- generate adjustment signals corresponding to selection and movement of said at least one adjustment point by said cursor,
- represent changes to the amplitude against frequency curve corresponding to the movement of said at least one adjustment point,
- adjust said filter characteristic response in accordance with said adjustment signals, and
- generate configuration signals to configure the configurable filter characteristic in accordance with said filter characteristic response.

2. An audio signal processing apparatus as claimed in Claim 1, comprising
- a data store which serves to store said filter characteristic response, and at least one other filter characteristic response.

3. An audio signal processing apparatus as claimed in Claim 1 or 2, wherein the configurable filter comprises a plurality of configurable filters, said filter characteristic response providing a characteristic response for each of said configurable filters and said curve of amplitude against frequency provides a curve of amplitude against frequency for each of said characteristic responses or a combined characteristic response.

4. An audio signal processing apparatus as claimed in Claims 1, 2 or 3, wherein each of said plurality of configurable filters is a digital filter, and said data processor operates to

- generate a plurality of filter coefficients corresponding to the filter characteristic response for each of said plurality of digital filters, said filter coefficients being represented by said configuration signals.

5. An audio signal processing apparatus as claimed in Claim 3 or 4, wherein the filter characteristic response for each of the configurable filters is at least one of a band pass filter, a low pass filter, a high pass filter or a shelf filter.

6. An audio signal processing apparatus as claimed in Claim 5, wherein said data processor operates to

- calculate said filter coefficients for each of said configurable filters from at least one of the gain at zero frequency, the gain at the Nyquist frequency, the gain at the resonant frequency and a warped frequency component.

7. An audio signal processing apparatus as claimed in Claims 6, wherein said plurality of configurable filters are infinite impulse response digital filters.

8. An audio signal processing apparatus as claimed in Claim 7, wherein said data processor operates to

- determine from said adjustments to said filter characteristic response which of the filter characteristic responses of said plurality of digital filters has changed,

- calculate new filter coefficients for each of the digital filters for which the filter characteristic response has changed,

- generate said configuration signals in accordance with the new filter coefficients. said configuration signals being used by said configurable filter to change the corresponding filter characteristics of the digital filters which have the new filter coefficients.

9. An audio signal processing apparatus as claimed in any of Claims 3 to 8, wherein said curve of amplitude against frequency has at least one of said adjustment points, each of which is selectable and moveable using said cursor, changes corresponding to the selection and movement of said adjustment points being represented by said data processor on said visual display means.

10. An audio signal processing apparatus as claimed in Claim 9, wherein said data processor operates to determine at least one of the gain at zero frequency, the gain at the Nyquist frequency, the gain at the resonant frequency and a warped frequency component following said movement of said adjustment points.

11. An audio signal processing apparatus as claimed in Claim 10, wherein said data processor operates to

- determine an effective frequency characteristic response for at least one warped half-effect frequency according to the calculated coefficients for at least one of the plurality of filters;

- determine an actual quality factor from the effective frequency characteristic response;

- re-calculate the gain in accordance with the actual quality factor; and

- re-calculate the filter coefficients from said re-calculated actual quality factor.

12. A method of processing audio signals comprising the steps of

- visually representing a curve of amplitude against frequency which is representative of a filter characteristic response;

- configuring a configurable filter in accordance with said filter characteristic response;
- filtering audio signals using said configurable filter;
- providing a visual representation of a cursor which is moveable in response to user generated commands;
- providing selectable and moveable adjustment points on said amplitude against frequency curve, said adjustment points being selectable and moveable in response to the position and movement of said cursor;
- adjusting said filter characteristic response in accordance with movement of said adjustment points effected by said cursor;
- visually representing the adjustment to said filter characteristic response; and
- re-configuring said configurable filter in accordance with the adjustment to said filter characteristic response.

13. A method of processing audio signals as claimed in Claim 12, comprising the step of

- storing said filter characteristic response and at least one other filter characteristic response
- selecting either said filter characteristic response or said other filter characteristic response to generate said amplitude against frequency curve and to configure said configurable filter.

14. A method of processing audio signals as claimed in Claim 12 or 13, wherein said filter comprises a plurality of configurable filters, said filter characteristic response comprises a characteristic response for each of said plurality of configurable filters, and said amplitude against frequency curve comprises a plurality of amplitude against frequency curves each of which is associated with a corresponding one of said plurality of filter characteristic responses, or a combined amplitude against frequency curve.

15. A method of processing audio signals as claimed in Claim 14, wherein each of said configurable filters is a digital filter, and said method comprises the step of

- calculating for each of said digital filters a set of filter coefficients representative of the corresponding characteristic filter response.

16. A method of processing audio signals as claimed in Claim 15 wherein the filter characteristic response for each of the configurable filters is at least one of a band pass filter, a low pass filter, a high pass filter or a shelf filter.

17. A method of processing audio signals as claimed in Claims 15 or 16, wherein said digital filters are infinite impulse response digital filters, said method including the steps of;

- determining at least one of the gain at zero frequency, the gain at the Nyquist frequency, the gain at the resonant frequency and a warped frequency component for each of the plurality of characteristic responses; and

- for each filter calculating said filter coefficients for each of said configurable filters from at least one of the gain at zero frequency, the gain at the Nyquist frequency, the gain at the resonant frequency and a warped frequency component.

18. A method of processing audio signals as claimed in any of Claims 13 to 17, wherein the step of re-configuring the configurable filter comprises the steps of

- determining which of the characteristic responses of said plurality of configurable filters has changed in response to said adjustment signals;

- calculating new filter coefficients for the configurable filters which have changed; and

- re-configuring those configurable filters for which the new filter coefficients have been calculated.

19. A method of processing audio signals as claimed in any of Claims 14 to 18, comprising the step of

- determining from adjusted curve at least one of the gain at zero frequency, the gain at the Nyquist frequency, the gain at the resonant frequency and a warped frequency component for the filter response.

20. A method a claimed in any of claims 16 to 19, comprising the steps of
- determining an effective frequency characteristic response for at least one warped half-effect frequency according to the calculated coefficients for at least one of the plurality of filters;
- determining an actual quality factor from the effective frequency characteristic response;
- re-calculating the gain in accordance with the actual quality factor; and
- re-calculating the filter coefficients from said re-calculated actual quality factor.

21. A computer program arranged to perform the steps of the method of processing audio signals as claimed in any of claims 12 to 20.

22. A computer readable carrier having stored thereon a computer program which is arranged to perform the steps of the method of processing audio signals as claimed in any of claims 12 to 20, when loaded on to a computer.

23. A computer operating as an audio signal processing apparatus having the features of any of claims 1 to 11, wherein the data processor operates in accordance with any of the steps of the method according to claims 12 to 20.

24. A programmable computer having an audio signal processing apparatus according to any of claims 1 to 11, wherein the computer when programmed performs any of the steps of the method of processing audio signals claimed in Claims 12 to 20.

25. An audio signal processing apparatus as herein before described with reference to the accompanying drawings.

26. A method of processing audio signals as herein before described with reference to the accompanying drawings.

The Patent Office

INVESTOR IN PEOPLE

Application No:	GB 9929446.4	**Examiner:**	Martyn Dixon
Claims searched:	all	**Date of search:**	6 June 2000

Patents Act 1977
Search Report under Section 17

Databases searched:

UK Patent Office collections, including GB, EP, WO & US patent specifications, in:

UK Cl (Ed.R): H4R (RPX,RSX); H4J (JGP,JGX)

Int Cl (Ed.7): H03G (5/02,5/16); H04H (7/00); H04S (7/00)

Other: Online: EPODOC,WPI,JAPIO

Documents considered to be relevant:

Category	Identity of document and relevant passage	Relevant to claims
A	GB 2330751 A (Sony)	
A	GB 2266210 A (Rodriguez)	
A	GB 2255696 A (Sony)	
A	US 5774566 A (Studer) see especially fig 3a	
A	US 4939782 A (Applied Research & Technology)	

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
		E	Patent document published on or after, but with priority date earlier than, the filing date of this application.
&	Member of the same patent family		